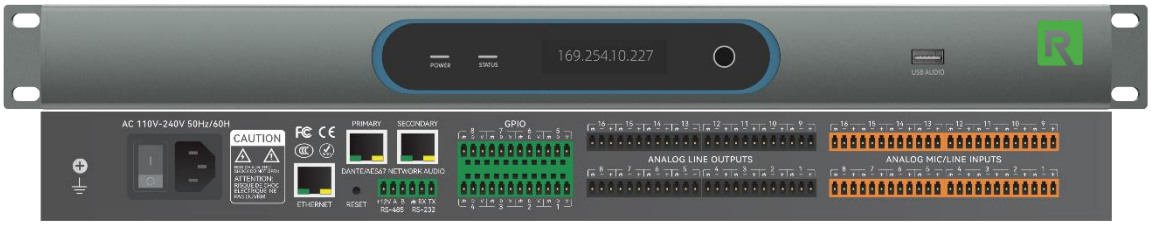
RESOUNDIFY

VoxNova1616

Open Structure DSP 16X16, Dante 64X64, 8AEC



PRODUCT OVERVIEW

The **Resoundify VoxNova1616** is a powerful and flexible open-architecture DSP solution designed for modern audio installations that demand high channel count, precise audio control, and seamless network integration. Featuring 16 analog inputs and 16 analog outputs, along with 64x64 Dante™ audio networking, it is ideal for conference rooms, multi-zone AV systems, and networked audio environments.

Built on a professional SHARC DSP platform, the VoxNova1616 provides 8 channels of full-duplex Acoustic Echo Cancellation (AEC), ensuring crystal-clear voice communication in even the most acoustically complex spaces. The open structure allows integrators to fully customize audio paths, filters, logic control, and DSP behavior using a graphical programming interface.

KEY FEATURES

- **Professional SHARC DSP Core:** Powered by Analog Devices' SHARC platform, the VoxNova1616 delivers ultra-fast signal processing with a semi-open architecture that supports complete customization of audio workflows and routing.
- **High-Quality Audio Processing:** 24-bit/48kHz audio resolution ensures crystal-clear sound quality across all channels.
- **Intelligent Feedback Suppression:** Independent adaptive feedback suppression on each channel automatically eliminates unwanted noise.
- **Full-Duplex AEC & ANC:** Integrated Adaptive Echo Cancellation and Active Noise Cancellation for clear communication in conferencing environments.
- **Auto Mixer & Gain Control:** Built-in Gain Sharing Auto Mixer, Automatic Gain Control (AGC), and Audio Ducking (Ducker) for seamless level balancing.
- **Ambient Noise Compensation:** Real-time Ambient Noise Compensator (ANC) adjusts audio levels based on environmental sound.
- **Comprehensive Audio Matrix:** Flexible mixing matrix with input level control, channel duplication, linking, and grouping.
- **Expandable Control Options:** 8 configurable GPIOs (input/output/ADC), RS-232 & UDP support with assignable ports for central control systems.
- **Multi-Platform Compatibility:** Supports both iOS and Windows OS with dual USB audio interface for recording and conferencing.

APPLICATIONS

- Boardrooms
- Classrooms
- Auditorium

RESOUNDIFY

TECHNICAL SPECIFICATIONS

System Specifications

Processor	ADI SHARC 21569@1GHz SIMD*2
Raw Processing Capacity	500 MIPS, 6 GFLOPS, 2 GMACS
Sampling Rate	48 kHz ± 100 ppm
Frequency Response (A/D/A)	20 Hz – 20 kHz ± 0.3 dB
Dynamic Range (A/D/A)	115 dB (A-weighted)
THD + Noise	<0.003%@4dBu
Channel Separation (A/D/A)	108 dB @ 1 kHz, +24 dBu
Latency (A/D/A)	<4 ms (input routed directly to output)
Delay Memory	174 mono seconds
Analog Control Inputs	0-3.3 VDC
Recommended External Control Potentiometer	10k Ohm, linear taper
Logic Outputs	Low (0 V) when active Pulled high (5 V) when inactive
Logic Output Maximum External Power Supply / Current Sinking	24 VDC / 50 mA
Logic Output Maximum Output Current	10 mA
RS-232 Accessory Serial I/O	57.6 kbps (default), 8 data bits, 1 stop bit, no parity, Straight-through wiring; pins 2, 3, 5 used
Maximum Stored Presets	1,000 storable presets

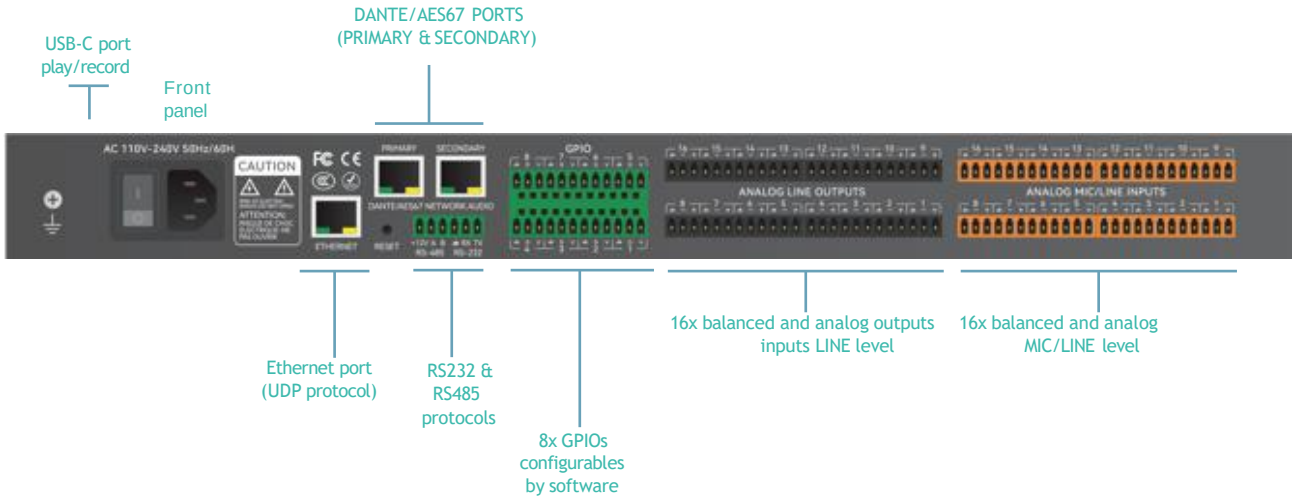
Analog Inputs and Outputs

Number of Analog Inputs	16 switchable balanced mic or line level
Analog Input and output Connectors	3.81 mm terminal blocks
Nominal Analog Input and output Level	+4 dBu with 20 dB headroom
Analog Input and output Maximum Level	+24 dBu (or +22.8 dBu into a 2k Ohm minimum load)
Analog Mic Pre-amp Gain	0 to 51 dB (in 3 dB steps) with ±24 dB digital trim
Analog Mic Pre-amp EIN	< -125 dB (with 150 Ohm source, 22.4 kHz BW)
Analog Input Impedance	5.4k Ohms balanced, 1k Ohms unbalanced
Analog Phantom Power (per input)	+48 VDC per input, max 10 mA
Analog Input Dynamic Range	>115 dB, A-weighted
Analog Input THD + Noise	<-100 dB (22.4 kHz BW, unweighted), 1 kHz @ +15 dBu, 0 dB gain
Analog Input Latency	2.5ms
Number of Analog Outputs	16 balanced line level
Analog Output Impedance	600 Ohms balanced, 300 Ohms unbalanced
Analog Output Dynamic Range	115 dB, A-weighted
Analog Output THD + Noise	< -97 dB (22.4 kHz BW, unweighted); 1 kHz, 0 dB gain, +8 dBu output
Analog Output Latency	1.5 ms

AEC8

AEC Number of Channels	8 Channels
AEC Tail Length	512 ms - suitable for medium rooms
AEC Convergence Rate	Typically > 90 dB/sec
AEC Latency	16 mS
AEC Processors	ADI SHARC 21569@1GHz

Rear View



Control Software

VoxControl+ is our dedicated configuration software, available for free download from our official website. Designed with a user-friendly interface, it allows fine-tunusers to easily tailor the matrix settings to match the specific needs of any installation. With this software, you can e a wide range of parameters, including:

- Input gain
- Expander
- Compressor & Limiter
- Auto Gain Control (AGC)
- Equalizer
- Figure Balancer
- Active Noise Control (ANC)
- Feedback (AFC)
- Noise gate
- Ducker
- SPL
- Share AM (Automixer)
- Echo Canceller (AEC)
- Camera Tracking
- Noise Suppresion (ANS)
- Matrix
- Low & High Pass filters
- Delayer
- Output